DEFENSE INFORMATION SYSTEMS AGENCY
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IN REPLY REFER TO: Joint Interoperability Test Command (JTE)

MEMORANDUM FOR DISTRIBUTION **29 Oct 10**

SUBJECT: Special Interoperability Test Certification of Microsoft Unified Communications Release v3.0.6362

References: (a) DoD Directive 4630.05, "Interoperability and Supportability of Information Technology (IT) and National Security Systems (NSS)," 5 May 2004
(b) CJCSI 6212.01E, "Interoperability and Supportability of Information Technology and National Security Systems," 15 December 2008
(c) through (h), see Enclosure 1

1. References (a) and (b) establish the Defense Information Systems Agency (DISA), Joint Interoperability Test Command (JITC), as the responsible organization for interoperability test certification.

2. The Microsoft Unified Communications Release v3.0.6362 is hereinafter referred to as the system under test (SUT). The SUT met all of the critical interoperability requirements and is certified for joint use within the Defense Switched Network (DSN) as a Private Branch Exchange 2 (PBX 2). The PBX 2 switches have no Military Unique Features (MUFs) and can only serve users having no requirement to originate Command and Control (C2) communications. Since PBX 2s do not support MUF Requirements detailed in Reference (c), connectivity to the DSN requires a waiver from the Chairman of the Joint Chiefs of Staff (CJCS) for each site in accordance with Reference (d). The SUT is certified for use with analog and Voice over Internet Protocol (VoIP) softphones (computers emulating telephones) only; the SUT was not tested and is not certified with VoIP hard phones (traditional desktop VoIP phones). The SUT meets the Voice over Internet Protocol critical interoperability requirements with any certified Assured Services Local Area Network (ASLAN) or non-ASLAN components on the Unified Capabilities (UC) Approved Products List (APL). The identified test discrepancies shown in the Certification Testing Summary (Enclosure 2) have been adjudicated as having an overall minor operational impact. No other configurations, features, or functions, except those cited within this report, are certified by the JITC. This certification expires upon changes that could affect interoperability, but no later than three years from the date of Defense Information Assurance (IA)/Security Accreditation Working Group (DSAWG) accreditation.

3. These findings are based on interoperability testing derived from Reference (e), DISA adjudication of open Test Discrepancy Reports (TDRs), review of the vendor's Letters of Compliance (LoC), and DSAWG accreditation. JITC completed interoperability testing of the SUT at the Global Information Grid Network Test Facility on 30 October 2009. Review of the vendor's LoC was completed on 8 December 2009. DISA completed adjudication of open TDRs

on 1 September 2010. Based upon DISA-led IA testing published separately in References (f) and (g), the DSAWG granted accreditation of the SUT and the supporting Microsoft Office Communicator Client Release 3.0.6362 on 21 October 2010 and 5 October 2010 respectively. Enclosure 2 documents the test results and describes the tested network and system configurations.

4. The interoperability test summary of the SUT is indicated in Table 1. The PBX 2 Capability Requirements (CRs) and Feature Requirements (FRs) are listed in Table 2. This interoperability test status is based on the SUT's ability to meet:

a. The DSN services for Network and Applications specified in Reference (d).

b. The PBX 2 interface and signaling requirements for trunks/lines specified in Reference (c) verified through JITC testing in accordance with Reference (e) and/or vendor submission of LoC.

c. The PBX 2 CRs/FRs specified in Reference (d) verified through JITC testing in accordance with Reference (e) and/or vendor submission of LoC.

d. The Internet Protocol CRs/FRs specified in References (c) and (h) verified through JITC testing in accordance with Reference (e) and/or vendor submission of LoC.

e. The softphone requirements specified in References (e) and (i).

f. The overall system interoperability performance derived from test procedures listed in Reference (e).

Table 1. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, MFR1, DP)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
T1 ISDN PRI NI 1/2 (ANSI T1.607)	Yes	Certified	Met all critical CRs and FRs.
E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with the following minor exceptions: The SUT 2-Wire analog interface is provided by their Audio Codes Mediant 1000 gateway. Due to interoperability interaction problems with line features supported by this gateway, the line features (i.e. call waiting, call hold, call transfer etc.) are disabled with a software patch and not authorized for use. Line features for a PBX 2 are not required; therefore the operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.

Table 1. SUT Interoperability Test Summary (continued)

DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
VoIP (Softphone only) (Ethernet IEEE 802.3u)	No	Certified	The SUT only supports softphones, it does not support VoIP hard phones. The SUT met all critical CRs and FRs with the following exception: During testing, there were two occasions when all softphones lost registration with the call controller and multiple occasions where individual softphones lost registration. When these failures occurred, the softphones logged out from the controller and lost active communications. Call processing was lost for a period of time, up to 10 minutes, with no known explanation, but did reregister and become active. DISA adjudicated this discrepancy as having a minor operational impact.
DSN Features and Capabilities			
Features and Capabilities	**Critical**	**Status**	**Remarks**
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exception: Due to interoperability interaction problems with line features supported by the Audio Codes Mediant 1000 gateway, the 2-Wire analog line features (i.e. call waiting, call hold, call transfer etc.) are disabled with a software patch and not authorized for use. The SUT supports line features on their softphones to include: transfer, call hold, 3 way conferencing, call waiting, and call forwarding. The SUT also supports other features not tested. There is no risk associated with not testing these other features supported by the SUT.
Public Safety	Yes	Certified	The SUT met the only required Public Safety requirement for a PBX 2: basic 911.
Call Processing	Yes	Certified	Met all critical CRs and FRs with following minor exception: During testing, there were two occasions when all softphones lost registration with the call controller and multiple occasions where individual softphones lost registration. When these failures occurred, the softphones logged out from the controller and lost active communications. Call processing was lost for a period of time, up to 10 minutes, with no known explanation, but did reregister and become active. DISA adjudicated this discrepancy as having a minor operational impact.
ISDN Services	No	Certified	ISDN Services are conditional for a PBX 2; however, the SUT offers an ISDN PRI interface and met the PRI Access, Call Control and Signaling requirements for this interface.
Synchronization	Yes	Certified	Met all critical CRs and FRs. The SUT meets the minimum requirement of line timing mode with their Audio Codes Mediant 1000 gateway which supports an internal clock of Stratum 4 or better.
Security	Yes	Certified	See note.
VoIP System	No	Certified	The SUT only supports softphones; it does not support VoIP hard phones. The SUT is certified for VoIP specifically with any certified ASLAN or non-ASLAN posted on the UC APL. In order to meet the Quality of Service requirements the SUT includes two Cisco Catalyst 3560G “edge” and “core” switches. The SUT is certified with these switches or any other layer 3 access switches listed on the UC APL.
Softphone	No	Certified	The SUT only supports softphones, it does not support VoIP hard phones. The SUT met all critical CRs and FRs with the following exception: During testing, there were two occasions when all softphones lost registration with the call controller and multiple occasions where individual softphones lost registration. When these failures occurred, the softphones logged out from the controller and lost active communications. Call processing was lost for a period of time, up to 10 minutes, with no known explanation, but did reregister and become active. DISA adjudicated this discrepancy as having a minor operational impact. Since the softphones do not provide tagging, they need to be connected directly to the Layer 3 switch, which will provide IEEE 802.1 p/Q VLAN tags, before connecting to a LAN as depicted in Enclosure 2, Figure 2-2.

Table 1. SUT Interoperability Test Summary (continued)

Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, DP)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	E1 CAS (DTMF, DP)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRS.
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRS.
	Ground Start Line	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.

NOTE: Security is tested by DISA-led Information Assurance test teams and published in separate reports, Reference (f) and (g).

LEGEND:

802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps
ANSI	American National Standards Institute
ASLAN	Assured Services Local Area Network
BRI	Basic Rate Interface
C2	Command and Control
CAS	Channel Associated Signaling
CRs	Capability Requirements
DISA	Defense Information Systems Agency
DP	Dial Pulse
DSN	Defense Switched Network
DSS1	Digital Subscriber Signaling 1
DTMF	Dual Tone Multi-Frequency
E1	European Basic Multiplex Rate (2.048 Mbps)
FRs	Feature Requirements
GR	Generic Requirement
GR-506-CORE	LSSGR: Signaling for Analog Interfaces
IEEE	Institute of Electrical and Electronics Engineers
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
Mbps	Megabits per second
MFR1	Multi-Frequency Recommendation 1
MLPP	Multi-Level Precedence and Preemption
MOS	Mean Opinion Score
NI 1/2	National ISDN Standard 1 or 2
PBX 2	Private Branch Exchange 2
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.931	Signaling Standard for ISDN
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
VLAN	Virtual Local Area Network
VoIP	Voice over Internet Protocol

Table 2. PBX 2 Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP) E1 CAS (MFR1, DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.607) E1 ISDN PRI (ITU-T Q.931)	No No (Europe only) No No (Europe only)	Trunking	• Direct Inward Dialing (C)	• UCR Section 5.2.1.3.2
			• National ISDN 1/2 Primary Access (C)	• UCR Section 5.2.1.3.4.1
			• ITU-T ISDN Primary Access (Europe only) (C)	• UCR Section 5.2.1.3.4.2
			• Normal Wink Start Operations (C)	• UCR Section 5.2.4.3.3.1.1
			• Glare Operation (C)	• UCR Section 5.2.4.3.3.1.2
			• Abnormal Wink Start (C)	• UCR Section 5.2.4.3.3.2
			• Glare Resolution (C)	• UCR Section 5.2.4.3.3.2.2
			• Call for Service Timing (C)	• UCR Section 5.2.4.3.5
			• Guard Timing (C)	• UCR Section 5.2.4.3.6
			• Satellite Interface (C)	• UCR Section 5.2.4.3.7
			• Disconnect Control (C)	• UCR Section 5.2.4.3.8
			• Reselect and Retrial (C)	• UCR Section 5.2.4.3.9
			• Off-Hook Supervision Transition (C)	• UCR Section 5.2.4.3.10
			• Dial-Pulse Signals (C)	• UCR Section 5.2.4.4.1
			• DTMF Signaling (C)	• UCR Section 5.2.4.4.2
			• DSN ISDN User-to-Network Signaling (C)	• UCR Section 5.2.4.7.1
			• Application (C)	• UCR Section 5.2.4.7.1.1
			• Physical Layer (C)	• UCR Section 5.2.4.7.1.2
			• Data Link Layer (C)	• UCR Section 5.2.4.7.1.3
			• Data Link Connection (C)	• UCR Section 5.2.4.7.1.3.1
			• Peer-to-Peer Procedures of Data-Link Layer (C)	• UCR Section 5.2.4.7.1.3.2
			• Layer 3 DSN User-to-Network Signaling (C)	• UCR Section 5.2.4.7.1.4
			• DSN User-to-Network Signaling for Circuit-Switched Bearer Services (C)	• UCR Section 5.2.4.7.1.4.2
			• Sequence of Messages for DSN Circuit-Switched Calls (C)	• UCR Section 5.2.4.7.1.4.3
			• Message Functional Definition and Content (C)	• UCR Section 5.2.4.7.1.4.4
			• General Message Format and Information Elements Coding (C)	• UCR Section 5.2.4.7.1.4.5
			• Supplementary Services (C)	• UCR Section 5.2.4.7.1.4.6
			• DSN Transmission Interface (C)	• UCR Section 5.2.5
			• PCM-24 Digital Trunk Interface (R)	• UCR Section 5.2.6.1
			• Interface Characteristics (R)	• UCR Section 5.2.6.1.1
			• Supervisory Channel Associated Signaling (C)	• UCR Section 5.2.6.1.2
			• Clear Channel Capability (C)	• UCR Section 5.2.6.1.3
			• Alarm and Restoral Requirements (C)	• UCR Section 5.2.6.1.4
			• PCM-30 Digital Trunk Interface (Europe only) (C)	• UCR Section 5.2.6.2
			• Interoperation of PCM-24 and PCM-30 (C)	• UCR Section 5.2.6.3
			• Analog Trunk Interface (C)	• UCR Section 5.2.6.4
			• Integrated Digital Loop Carrier (C)	• UCR Section 5.2.6.5
		Voice	• MOS (R)	• CJCSI 6215.01C
			• Secure calls (C)	• CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R)	• CJCSI 6215.01C
			• 56 kbps switched data (C: PRI only)	• UCR Section 5.2.2.9.6
			• 64 kbps switched data (C: PRI only)	• UCR Section 5.2.2.9.6
			• NX56 synchronous BER (C: PRI only)	• UCR Section 5.2.2.9.6
			• NX64 synchronous BER (C: PRI only)	• UCR Section 5.2.2.9.6
			• Secure data (STE/STU-III) (C)	• CJCSI 6215.01C
		VTC	• ITU-T H.320 (C: PRI only)	• FTR 1080B-2002

Table 2. PBX 2 Requirements (continued)

DSN Line Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
2-Wire Analog ISDN BRI NI 1/2 2-Wire Proprietary Digital	Yes No No	Access	• Individual Line (R) • PBX Line (C) • National ISDN 1/2 Basic Access (C) • Analog Line (C) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (C) • S/T Reference Point (C: ISDN BRI only)	• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.3.1 • UCR Section 5.2.1.3.3 • UCR Section 5.2.1.3.5 • UCR Section 5.2.4.2.1 • UCR Section 5.2.4.3.1 • UCR Section 5.2.4.7.1.2.1
		Voice	• MOS (R) • Secure Calls (C)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R) • Secure data (STE/STU-III) (C)	• CJCSI 6215.01C • CJCSI 6215.01C
		VTC	• ITU-T H.320 (C: BRI only)	• FTR 1080B-2002

DSN Features & Capabilities			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
Common Features	Yes	• Individual Lines (R) • Call Waiting (C) • Three-way Calling (C) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (C) • Call Transfer - Internal Only (C) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (C) • Call Transfer – Outside (C) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (C) • Conference Calling – Six Way Station Controlled (C) • Call Forwarding Variable (C) • Call Forward Busy Line (C) • Call Forwarding – Don't Answer – All Calls (C) • Selective Call Forwarding (C) • Call pick-up (C)	• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.1.5.1 • UCR Section 5.2.1.1.6 • UCR Section 5.2.1.1.7 • UCR Section 5.2.1.1.7.1 • UCR Section 5.2.1.1.7.2 • UCR Section 5.2.1.1.7.3 • UCR Section 5.2.1.1.7.4 • UCR Section 5.2.1.1.7.5 • UCR Section 5.2.1.1.7.6 • UCR Section 5.2.1.1.7.7 • UCR Section 5.2.1.1.7.8 • UCR Section 5.2.1.1.8.1 • UCR Section 5.2.1.1.8.2 • UCR Section 5.2.1.1.8.3 • UCR Section 5.2.1.1.8.4 • UCR Section 5.2.1.1.9.1
Public Safety	Yes	• Emergency Service (911) Caller (R) • Emergency Service (911) Public Safety Answering Point (C) • Enhanced Emergency Service (E911) (C)	• UCR Section 5.2.1.4.1.1 • UCR Section 5.2.1.4.1.2 • UCR Section 5.2.1.4.1.3
Call Processing	Yes	• Origination Treatment (R) • Originating Busy (R) • Termination Treatment (R) • Busy or Idle Status (C) • Release Treatment (R) • Interruption Treatment (R) • Connections (R) • Class of Service (C) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (C) • Interswitch and Intraswitch Dialing (C) • Calling Name Delivery (C) • Calling Number Delivery (C) • Screening (C)	• UCR Section 5.2.3.1.1 • UCR Section 5.2.3.1.1.1 • UCR Section 5.2.3.1.2 • UCR Section 5.2.3.1.2.1 • UCR Section 5.2.3.1.3 • UCR Section 5.2.3.1.4 • UCR Section 5.2.3.1.5 • UCR Section 5.2.3.1.6 • UCR Section 5.2.3.3.1 • UCR Section 5.2.3.3.2 • UCR Section 5.2.3.3.3 • UCR Section 5.2.3.5.1.2 • UCR Section 5.2.3.5.1.8.1 • UCR Section 5.2.3.5.1.8.2 • UCR Section 5.2.3.5.8

Table 2. PBX 2 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
ISDN Services	No	• BRI Access, Call Control and Signaling (C) • Uniform Interface Configuration for BRIs (C) • BRI Features (C) • PRI Access, Call Control and Signaling (C) • PRI Features (C) • Packet Data Features and Capabilities (C)	• UCR Section 5.2.9.2 Table5.2.9.-1 • UCR Section 5.2.9.2 Table 5.2.9-2 • UCR Section 5.2.9.2 Table 5.2.9-3 • UCR Section 5.2.9.2 Table 5.2.9-4 • UCR Section 5.2.9.2 Table 5.2.9-5 • UCR Section 5.2.9.2 Table 5.2.9-6
Synchronization	Yes	• Line timing mode (C) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 5.2.10.1.1.2 • UCR Section 5.2.10.1.2.2 • UCR Section 5.2.10.2 • UCR Section 5.2.10.3 • UCR Section 5.2.10.4
VoIP System		• Voice Quality with MOS of 4.0 or better IAW 5.3.3.14 (R) (IP Softphones exempt) • ITU-T G.711 PCM CODEC (R) • MLPP (C) • Security (R) • Network management (C) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging IAW 5.3.1 (R) • Packet Loss	• UCR Section 5.2.12.8.2.1 • UCR Section 5.2.12.8.2.2 • UCR Section 5.2.12.8.2.3 • UCR Section 5.2.12.8.2.4 • UCR Section 5.2.12.8.2.5 • UCR Section 5.2.12.8.2.6 • UCR Section 5.2.12.8.2.7 • UCR Section 5.2.12.8.2.8 • UCR Section 5.2.12.8.2.9 • UCR Section 5.3.1.3
Softphone		• Softphone Requirements (R)	• DISA Memo (Reference h)
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Sections 3 and 5.4

Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (C) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.4.2.2 • UCR Section 5.2.4.3.2 • UCR Section 5.2.4.3.4

Table 2. PBX 2 Requirements (continued)

LEGEND:

ANSI	American National Standards Institute
BER	Bit Error Ratio
BRI	Basic Rate Interface
C	Conditional
CAS	Channel Associated Signaling
CJCSI	Chairman of the Joint Chiefs of Staff Instruction
DIACAP	DoD Information Assurance Certification and Accreditation Process
DISR	DoD IT Standards Registry
DoD	Department of Defense
DoDI	DoD Instruction
DP	Dial Pulse
DS0	Digital Signal Level 0
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)
DSN	Defense Switched Network
DSS1	Digital Subscriber Signaling 1
DTMF	Dual Tone Multi-Frequency
E1	European Basic Multiplex Rate (2.048 Mbps)
E911	Enhanced 911 Service
E&M	Ear and Mouth
FTR	Federal Telecommunications Recommendation
FTR 1080B-2002	Video Teleconferencing Services
G.711	PCM of voice frequencies
GR	Generic Requirement
GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security
H.320	Standard for Narrowband VTC
IAW	in accordance with
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
IT	Information Technology
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
kbps	kilobits per second
Mbps	Megabits per second
MFR1	Multi-Frequency Recommendation 1
MLPP	Multi-Level Precedence and Preemption
MOS	Mean Opinion Score
NI 1/2	National ISDN Standard 1 or 2
NX56	Data format restricted to multiples of 56 kbps
NX64	Data format restricted to multiples of 64 kbps
PBX	Private Branch Exchange
PBX 2	Private Branch Exchange 2
PCM-24	Pulse Code Modulation - 24 Channels
PCM-30	Pulse Code Modulation - 30 Channels
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.931	Signaling Standard for ISDN
R	Required
S/T	ISDN BRI 4-wire interface
STE	Secure Terminal Equipment
STIGs	Security Technical Implementation Guides
STU-III	Secure Telephone Unit -3rd generation
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
T.4	Standardization of Group 3 facsimile terminals for document transmission
UCR	Unified Capabilities Requirements
VBD	Variable bit data
VoIP	Voice over Internet Protocol
VTC	Video Teleconferencing

5. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: ucco@disa.mil.

6. The JITC point of contact is Mr. Edward Mellon, DSN 879-5159, commercial (520) 538-5159, FAX DSN 879-4347, or e-mail to edward.mellon@disa.mil. The JITC's mailing address is P.O. Box 12798, Fort Huachuca, AZ 85670-2798. The tracking number for the SUT is 0913302. The tracking number for the Microsoft Office Communicator is 0918003.

FOR THE COMMANDER:

RAMeador

2 Enclosures a/s

RICHARD A. MEADOR
Chief
Battlespace Communications Portfolio

Distribution (electronic mail):
Joint Staff J-6
Joint Interoperability Test Command, Liaison, TE3/JT1
Office of Chief of Naval Operations, CNO N6F2
Headquarters U.S. Air Force, Office of Warfighting Integration & CIO, AF/XCIN (A6N)
Department of the Army, Office of the Secretary of the Army, DA-OSA CIO/G-6 ASA (ALT), SAIS-IOQ
U.S. Marine Corps MARCORSYSCOM, SIAT, MJI Division I
DOT&E, Net-Centric Systems and Naval Warfare
U.S. Coast Guard, CG-64
Defense Intelligence Agency
National Security Agency, DT
Defense Information Systems Agency, TEMC
Office of Assistant Secretary of Defense (NII)/DOD CIO
U.S. Joint Forces Command, Net-Centric Integration, Communication, and Capabilities Division, J68
Defense Information Systems Agency, GS23

ADDITIONAL REFERENCES

(c) Office of the Assistant Secretary of Defense, "Department of Defense Unified Capabilities Requirements 2008," 22 January 2009
(d) Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)," 9 November 2007
(e) Joint Interoperability Test Command, "Defense Switched Network Generic Switch Test Plan (GSTP), Change 2," 2 October 2006
(f) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Microsoft Unified Communications, Release (Rel.) version (v)3.0.6362 (TN0913302)," 21 October 2010
(g) Joint Interoperability Test Command, "Information Assurance (IA) Assessment of Microsoft Office Communicator Client Release (Rel.) version (v) 3.0.6362 (Tracking Number 0918003)," 5 October 2010
(h) Office of the Secretary of Defense, "Interim Unified Capabilities (UC) IPv6 Rules of Engagement (ROE)," 31 July 2009
(i) Defense Information Systems Agency NS3 Memorandum, "Softphone Certification" 20 April 2009

CERTIFICATION TESTING SUMMARY

1. SYSTEM TITLE. Microsoft Unified Communications Release v3.0.6362; hereinafter referred to as the System Under Test (SUT).

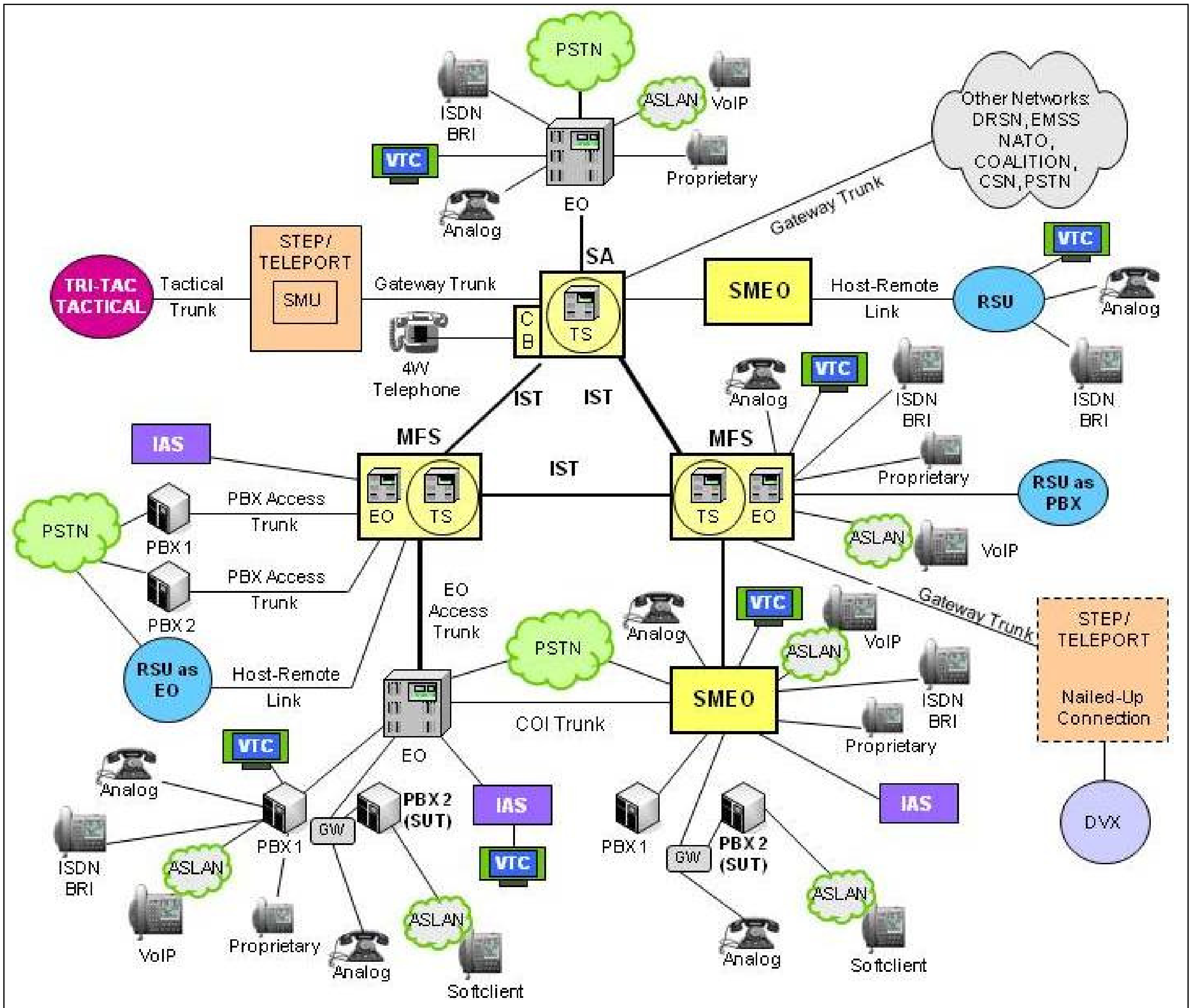
2. PROPONENT. Program Manager Defense Communications and Switched Systems, Technical Management Division (PM DCASS-TMD).

3. PROGRAM MANAGER. Miguel S. Buddle, SFAE-PS-SW-T, 283 Sherril Avenue, Fort Monmouth, New Jersey, 07703, e-mail: Miguel.s.buddle@us.army.mil.

4. TESTER. Joint Interoperability Test Command (JITC), Fort Huachuca, Arizona.

5. SYSTEM UNDER TEST DESCRIPTION. The SUT is a Voice over Internet Protocol (VoIP) approach to the Private Branch Exchange 2 (PBX 2) with voice mail. The SUT includes multiple servers running Windows Server 2003 Service Pack (SP)2 operating system and Office Communication Server (OCS) 2007 software. The Front End (FE) is the server component, which does the call control role that supplies Session Initiation Protocol (SIP) communications between end points, Audio/Video (A/V) and web conferencing functionality, and Real-Time Transport Protocol (RTP) through the perimeter network. The Back End (BE) server is the real-time data store for state information and is based on Microsoft (MS) Structured Query Language (SQL). The OCS-Mediation (MED)1 server is used for Codec compatibility with legacy systems and gateways. The Domain Controller (DC), which should already be on site, is used for authentication. The Exchange 2007 SP1 is used for access, storage, and encryption for sending and receiving of all voice mail traffic. The Unified Communications System registers the endpoints and the Microsoft Office Communicator Client SoftPhone on the workstations. The Cisco switches are used for Virtual Local Area Network (VLAN) tagging, and quality of service layer 3 tagging. The F5 Load Balancers are used for signaling and media availability, load balancing failover, and load distribution. The Unified Communications System uses a Time Division Multiplexing (TDM)/Internet Protocol (IP) media gateway, AudioCodes Mediant 1000, to convert IP signaling and media to connect to legacy protocols for Defense Switched Network (DSN) and Public Switched Telephone Network (PSTN) connectivity. The SUT was tested and is certified for use with analog and VoIP softphones (computers emulating telephones) only. The SUT was not tested and is not certified with VoIP hard phones (traditional desktop VoIP phones).

6. OPERATIONAL ARCHITECTURE. The Defense Switched Network (DSN) architecture is a two-level network hierarchy consisting of DSN backbone switches and Service/Agency installation switches. Joint Staff policy and subscriber mission requirements determine which type of switch can be used at a particular location. The DSN architecture, therefore, consists of several categories of switches including PBXs. The Unified Capabilities Requirements (UCR) operational DSN Architecture is depicted in Figure 2-1. The architecture depicts the relationship of Military Department PBX 1s to the other DSN switch types.



LEGEND:

4W	4-Wire	NATO	North Atlantic Treaty Organization
ASLAN	Assured Services Local Area Network	PBX	Private Branch Exchange
BRI	Basic Rate Interface	PBX 1	Private Branch Exchange 1
CB	Channel Bank	PBX 2	Private Branch Exchange 2
COI	Community of Interest	PSTN	Public Switched Telephone Network
CSN	Canadian Switch Network	RSU	Remote Switching Unit
DRSN	Defense Red Switch Network	SA	Standalone
DSN	Defense Switched Network	SMEO	Small End Office
DVX	Deployable Voice Exchange	SMU	Switched Multiplex Unit
EMSS	Enhanced Mobile Satellite System	STEP	Standardized Tactical Entry Point
EO	End Office	SUT	System Under Test
GW	Gateway	Tri-Tac	Tri-Service Tactical Communications Program
IAS	Integrated Access Switch	TS	Tandem Switch
ISDN	Integrated Services Digital Network	VoIP	Voice over Internet Protocol
IST	Interswitch Trunk	VTC	Video Teleconferencing
MFS	Multifunction Switch		

Figure 2-1. DSN Architecture

7. REQUIRED SYSTEM INTERFACES. Requirements specific to PBX 2s are listed in Table 2-1. These requirements are derived from:

a. The DSN services for Network and Applications specified in Chairman of the Joint Chiefs of Staff Instruction (CJCSI) 6215.01C, "Policy for Department of Defense Voice Services with Real Time Services (RTS)", Reference (d).

b. The PBX 2 interface and signaling requirements for trunks/lines specified in Reference (c) verified through JITC testing in accordance with Reference (e) and/or vendor submission of LoC.

c. The PBX 2 CRs/FRs specified in Reference (d) verified through JITC testing in accordance with Reference (e) and/or vendor submission of LoC.

d. The Internet Protocol CRs/FRs specified in References (c) and (h) verified through JITC testing in accordance with Reference (e) and/or vendor submission of LoC.

e. The softphone requirements specified in References (e) and (i).

Table 2-1. PBX 2 Requirements

DSN Trunk Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
T1 CAS (MFR1, DTMF, DP) E1 CAS (MFR1, DTMF, DP) T1 ISDN PRI NI 1/2 (ANSI T1.607) E1 ISDN PRI (ITU-T Q.931)	No No (Europe only) No No (Europe only)	Trunking	• Direct Inward Dialing (C)	• UCR Section 5.2.1.3.2
			• National ISDN 1/2 Primary Access (C)	• UCR Section 5.2.1.3.4.1
			• ITU-T ISDN Primary Access (Europe only) (C)	• UCR Section 5.2.1.3.4.2
			• Normal Wink Start Operations (C)	• UCR Section 5.2.4.3.3.1.1
			• Glare Operation (C)	• UCR Section 5.2.4.3.3.1.2
			• Abnormal Wink Start (C)	• UCR Section 5.2.4.3.3.2
			• Glare Resolution (C)	• UCR Section 5.2.4.3.3.2.2
			• Call for Service Timing (C)	• UCR Section 5.2.4.3.5
			• Guard Timing (C)	• UCR Section 5.2.4.3.6
			• Satellite Interface (C)	• UCR Section 5.2.4.3.7
			• Disconnect Control (C)	• UCR Section 5.2.4.3.8
			• Reselect and Retrial (C)	• UCR Section 5.2.4.3.9
			• Off-Hook Supervision Transition (C)	• UCR Section 5.2.4.3.10
			• Dial-Pulse Signals (C)	• UCR Section 5.2.4.4.1
			• DTMF Signaling (C)	• UCR Section 5.2.4.4.2
			• DSN ISDN User-to-Network Signaling (C)	• UCR Section 5.2.4.7.1
			• Application (C)	• UCR Section 5.2.4.7.1.1
			• Physical Layer (C)	• UCR Section 5.2.4.7.1.2
			• Data Link Layer (C)	• UCR Section 5.2.4.7.1.3
			• Data Link Connection (C)	• UCR Section 5.2.4.7.1.3.1
			• Peer-to-Peer Procedures of Data-Link Layer (C)	• UCR Section 5.2.4.7.1.3.2
			• Layer 3 DSN User-to-Network Signaling (C)	• UCR Section 5.2.4.7.1.4
			• DSN User-to-Network Signaling for Circuit-Switched Bearer Services (C)	• UCR Section 5.2.4.7.1.4.2
			• Sequence of Messages for DSN Circuit-Switched Calls (C)	• UCR Section 5.2.4.7.1.4.3
			• Message Functional Definition and Content (C)	• UCR Section 5.2.4.7.1.4.4
			• General Message Format and Information Elements Coding (C)	• UCR Section 5.2.4.7.1.4.5
			• Supplementary Services (C)	• UCR Section 5.2.4.7.1.4.6
			• DSN Transmission Interface (C)	• UCR Section 5.2.5
			• PCM-24 Digital Trunk Interface (R)	• UCR Section 5.2.6.1
			• Interface Characteristics (R)	• UCR Section 5.2.6.1.1
			• Supervisory Channel Associated Signaling (C)	• UCR Section 5.2.6.1.2
			• Clear Channel Capability (C)	• UCR Section 5.2.6.1.3
			• Alarm and Restoral Requirements (C)	• UCR Section 5.2.6.1.4
			• PCM-30 Digital Trunk Interface (Europe only) (C)	• UCR Section 5.2.6.2
			• Interoperation of PCM-24 and PCM-30 (C)	• UCR Section 5.2.6.3
			• Analog Trunk Interface (C)	• UCR Section 5.2.6.4
			• Integrated Digital Loop Carrier (C)	• UCR Section 5.2.6.5
		Voice	• MOS (R)	• CJCSI 6215.01C
			• Secure calls (C)	• CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R)	• CJCSI 6215.01C
			• 56 kbps switched data (C: PRI only)	• UCR Section 5.2.2.9.6
			• 64 kbps switched data (C: PRI only)	• UCR Section 5.2.2.9.6
			• NX56 synchronous BER (C: PRI only)	• UCR Section 5.2.2.9.6
			• NX64 synchronous BER (C: PRI only)	• UCR Section 5.2.2.9.6
			• Secure data (STE/STU-III) (C)	• CJCSI 6215.01C
		VTC	• ITU-T H.320 (C: PRI only)	• FTR 1080B-2002

Table 2-1. PBX 2 Requirements (continued)

DSN Line Interfaces				
Interface	**Critical**	**Requirements Required or Conditional**		**References**
2-Wire Analog ISDN BRI NI 1/2 2-Wire Proprietary Digital	Yes No No	Access	• Individual Line (R) • PBX Line (C) • National ISDN 1/2 Basic Access (C) • Analog Line (C) • Loop Start Line (R: 2-Wire Analog only) • Reverse Battery (C) • S/T Reference Point (C: ISDN BRI only)	• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.3.1 • UCR Section 5.2.1.3.3 • UCR Section 5.2.1.3.5 • UCR Section 5.2.4.2.1 • UCR Section 5.2.4.3.1 • UCR Section 5.2.4.7.1.2.1
		Voice	• MOS (R) • Secure Calls (C)	• CJCSI 6215.01C • CJCSI 6215.01C
		Facsimile	• Analog: ITU-T T.4 (R)	• DISR
		Data	• Modem (VBD) (R) • Secure data (STE/STU-III) (C)	• CJCSI 6215.01C • CJCSI 6215.01C
		VTC	• ITU-T H.320 (C: BRI only)	• FTR 1080B-2002
DSN Features & Capabilities				
Feature/ Capability	**Critical**	**Requirements Required or Conditional**		**References**
Common Features	Yes	• Individual Lines (R) • Call Waiting (C) • Three-way Calling (C) • Add-on transfer, conference calling, and call hold (C) • Call Transfer Individual – All calls (C) • Call Transfer - Internal Only (C) • Call Transfer – Individual – Incoming Only/Add-On Consultation Hold – Incoming Call (C) • Call Transfer – Outside (C) • Call Transfer – Add-On Restricted Station (C) • Call Transfer – Attendant (C) • Call Hold (C) • Conference Calling – Six Way Station Controlled (C) • Call Forwarding Variable (C) • Call Forward Busy Line (C) • Call Forwarding – Don't Answer – All Calls (C) • Selective Call Forwarding (C) • Call pick-up (C)		• UCR Section 5.2.1.1.1 • UCR Section 5.2.1.1.5.1 • UCR Section 5.2.1.1.6 • UCR Section 5.2.1.1.7 • UCR Section 5.2.1.1.7.1 • UCR Section 5.2.1.1.7.2 • UCR Section 5.2.1.1.7.3 • UCR Section 5.2.1.1.7.4 • UCR Section 5.2.1.1.7.5 • UCR Section 5.2.1.1.7.6 • UCR Section 5.2.1.1.7.7 • UCR Section 5.2.1.1.7.8 • UCR Section 5.2.1.1.8.1 • UCR Section 5.2.1.1.8.2 • UCR Section 5.2.1.1.8.3 • UCR Section 5.2.1.1.8.4 • UCR Section 5.2.1.1.9.1
Public Safety	Yes	• Emergency Service (911) Caller (R) • Emergency Service (911) Public Safety Answering Point (C) • Enhanced Emergency Service (E911) (C)		• UCR Section 5.2.1.4.1.1 • UCR Section 5.2.1.4.1.2 • UCR Section 5.2.1.4.1.3
Call Processing	Yes	• Origination Treatment (R) • Originating Busy (R) • Termination Treatment (R) • Busy or Idle Status (C) • Release Treatment (R) • Interruption Treatment (R) • Connections (R) • Class of Service (C) • E&M Lead Signaling States (C) • 4-Wire Analog User Access Lines (C) • 2-Wire User Access Lines (C) • Interswitch and Intraswitch Dialing (C) • Calling Name Delivery (C) • Calling Number Delivery (C) • Screening (C)		• UCR Section 5.2.3.1.1 • UCR Section 5.2.3.1.1.1 • UCR Section 5.2.3.1.2 • UCR Section 5.2.3.1.2.1 • UCR Section 5.2.3.1.3 • UCR Section 5.2.3.1.4 • UCR Section 5.2.3.1.5 • UCR Section 5.2.3.1.6 • UCR Section 5.2.3.3.1 • UCR Section 5.2.3.3.2 • UCR Section 5.2.3.3.3 • UCR Section 5.2.3.5.1.2 • UCR Section 5.2.3.5.1.8.1 • UCR Section 5.2.3.5.1.8.2 • UCR Section 5.2.3.5.8

Table 2-1. PBX 2 Requirements (continued)

DSN Features & Capabilities (continued)			
Feature/ Capability	**Critical**	**Requirements Required or Conditional**	**References**
ISDN Services	No	• BRI Access, Call Control and Signaling (C) • Uniform Interface Configuration for BRIs (C) • BRI Features (C) • PRI Access, Call Control and Signaling (C) • PRI Features (C) • Packet Data Features and Capabilities (C)	• UCR Section 5.2.9.2 Table5.2.9.-1 • UCR Section 5.2.9.2 Table 5.2.9-2 • UCR Section 5.2.9.2 Table 5.2.9-3 • UCR Section 5.2.9.2 Table 5.2.9-4 • UCR Section 5.2.9.2 Table 5.2.9-5 • UCR Section 5.2.9.2 Table 5.2.9-6
Synchronization	Yes	• Line timing mode (C) • Internal Stratum 4 (R) • Synchronization Performance Monitoring Criteria (C) • DS1 Traffic Interfaces (C) • DS0 Traffic Interconnects (C)	• UCR Section 5.2.10.1.1.2 • UCR Section 5.2.10.1.2.2 • UCR Section 5.2.10.2 • UCR Section 5.2.10.3 • UCR Section 5.2.10.4
VoIP System		• Voice Quality with MOS of 4.0 or better IAW 5.3.3.14 (R) (IP Softphones exempt) • ITU-T G.711 PCM CODEC (R) • MLPP (C) • Security (R) • Network management (C) • System timing (R) • Latency ≤ 60 milliseconds (R) • IPv6 capable (R) • Service Class Tagging IAW 5.3.1 (R) • Packet Loss	• UCR Section 5.2.12.8.2.1 • UCR Section 5.2.12.8.2.2 • UCR Section 5.2.12.8.2.3 • UCR Section 5.2.12.8.2.4 • UCR Section 5.2.12.8.2.5 • UCR Section 5.2.12.8.2.6 • UCR Section 5.2.12.8.2.7 • UCR Section 5.2.12.8.2.8 • UCR Section 5.2.12.8.2.9 • UCR Section 5.3.1.3
Softphone		• Softphone Requirements (R)	• DISA Memo (Reference h)
Security	Yes	• GR-815, STIGs, and DoDI 8510.bb (DIACAP) (R)	• UCR Sections 3 and 5.4

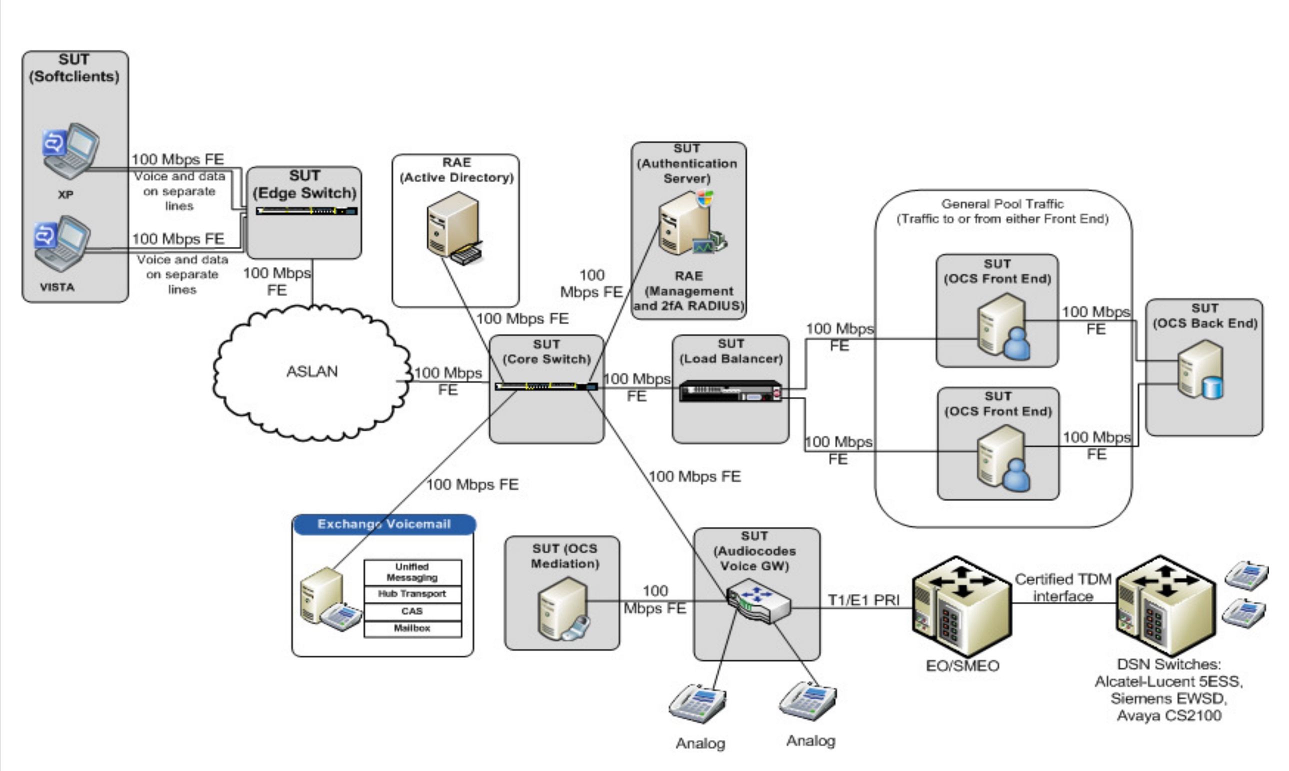
Network Gateways				
Gateway	**Critical**	**Requirements Required or Conditional**		**References**
PSTN	No	Trunking	• Positive Identification Control (C) • On-Netting (C) • Off-Netting (C) • Ground Start Line (C) • Immediate Start (C) • Delay Dial (C)	• CJCSI 6215.01C • CJCSI 6215.01C • CJCSI 6215.01C • UCR Section 5.2.4.2.2 • UCR Section 5.2.4.3.2 • UCR Section 5.2.4.3.4

Table 2-1. PBX 2 Requirements (continued)

LEGEND:

ANSI	American National Standards Institute
BER	Bit Error Ratio
BRI	Basic Rate Interface
C	Conditional
CAS	Channel Associated Signaling
CJCSI	Chairman of the Joint Chiefs of Staff Instruction
DIACAP	DoD Information Assurance Certification and Accreditation Process
DISR	DoD IT Standards Registry
DoD	Department of Defense
DoDI	DoD Instruction
DP	Dial Pulse
DS0	Digital Signal Level 0
DS1	Digital Signal Level 1 (1.544 Mbps) (2.048 Mbps European)
DSN	Defense Switched Network
DSS1	Digital Subscriber Signaling 1
DTMF	Dual Tone Multi-Frequency
E1	European Basic Multiplex Rate (2.048 Mbps)
E911	Enhanced 911 Service
E&M	Ear and Mouth
FTR	Federal Telecommunications Recommendation
FTR 1080B-2002	Video Teleconferencing Services
G.711	PCM of voice frequencies
GR	Generic Requirement
GR-815	Generic Requirements For Network Element/Network System (NE/NS) Security
H.320	Standard for Narrowband VTC
IAW	in accordance with
IPv6	Internet Protocol version 6
ISDN	Integrated Services Digital Network
IT	Information Technology
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
kbps	kilobits per second
Mbps	Megabits per second
MFR1	Multi-Frequency Recommendation 1
MLPP	Multi-Level Precedence and Preemption
MOS	Mean Opinion Score
NI 1/2	National ISDN Standard 1 or 2
NX56	Data format restricted to multiples of 56 kbps
NX64	Data format restricted to multiples of 64 kbps
PBX	Private Branch Exchange
PBX 2	Private Branch Exchange 2
PCM-24	Pulse Code Modulation - 24 Channels
PCM-30	Pulse Code Modulation - 30 Channels
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.931	Signaling Standard for ISDN
R	Required
S/T	ISDN BRI 4-wire interface
STE	Secure Terminal Equipment
STIGs	Security Technical Implementation Guides
STU-III	Secure Telephone Unit -3rd generation
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
T.4	Standardization of Group 3 facsimile terminals for document transmission
UCR	Unified Capabilities Requirements
VBD	Variable bit data
VoIP	Voice over Internet Protocol
VTC	Video Teleconferencing

8. TEST NETWORK DESCRIPTION. The SUT was tested at JITC’s Global Information Grid Network Test Facility in a manner and configuration similar to that of the DSN operational environment. Testing of the system’s required functions and features was conducted using test configuration depicted in Figure 2-2. The SUT was tested as the end-point in relation to the other switches.



LEGEND:

2fA	Two factor Authentication	Mbps	Megabits per second
5ESS	Class 5 Electronic Switching System	OCS	Office Communication Server
ASLAN	Assured Services Local Area Network	PRI	Primary Rate Interface
CS	Communication Server	RADIUS	Remote Authentication Dial In User Service
DSN	Defense Switched Network	RAE	Required Ancillary Equipment
E1	European Basic Multiplex Rate (2.048 Mbps)	SMEO	Small End Office
EO	End Office	SUT	System Under Test
EWSD	Elektronisches Wählsystem Digital	T1	Digital Transmission Link Level 1 (1.544 Mbps)
FE	Fast Ethernet	TDM	Time Division Multiplexing
GW	Gateway		

Figure 2-2. SUT Test Configuration

9. SYSTEM CONFIGURATIONS. Table 2-2 provides the system configurations, hardware, and software components tested with the SUT. The SUT was tested in an operationally realistic environment to determine interoperability with a complement of DSN switches noted in Table 2-2. Table 2-2 lists the DSN switches which depict the tested configuration and is not intended to identify the only switches that are certified with the SUT. The SUT is certified with switching systems listed on the Unified Capabilities (UC) Approved Products List (APL) that offer the same certified interfaces.

Table 2-2. Tested System Configurations

System Name	Software Release
Avaya CS2100	Succession Enterprise (SE) 09.1
Nokia-Siemens EWSD	19d with Patch Set 46
Avaya S8720	Communication Manager (CM) 4.0 (R014x.00.2.732.1: Super Patch 16538)
Alcatel-Lucent 5ESS	5E16.2 BWM 09-0002

System Name	Hardware/Software Release
Required Ancillary Equipment	Active Directory
	Public Key Infrastructure
	Remote Authentication Dial in User Service
	SysLog Server

System Name	Hardware	Card Name Part Number/ Name	Software/ Firmware
Microsoft Unified Communications, Rel. v3.0.6362	Windows XP Workstation	NA	Microsoft Windows XP SP3
			Office Communicator Client Rel. v3.56907.83
			Microsoft .NET Framework 3.5 SP1
			Tumbleweed 4.10.0.344
			ActivClient CAC 6.1 x86
	Windows Vista Workstation	NA	Microsoft Windows Vista SP1
			Office Communicator Client Rel. v3.56907.83
			Microsoft .NET Framework 3.5 SP1
			Tumbleweed 4.10.0.344
			ActivClient CAC 6.1 x86
	Cisco Edge Switch	Catalyst 3560G	IOS 12.2.(25) SEE4
	Cisco Core Switch	Catalyst 3560G	IOS 12.2.(25) SEE4
	F5 Load Balancer	bip251597s	Big-IP6400 v10.01 build 354.0Hotfix HF2
	AuthSer	HP Proliant BL460c	Microsoft Windows 2003 SP2
			IAS Windows 2003 SP2
			ISA (RADIUS) 2006 SP1 v5.0.5723.493
			Tumbleweed 4.10.0.344
			ActivClient CAC 6.1 x86

Table 2-2. Tested System Configurations (continued)

System Name	Hardware	Card Name / Part Number/ Name	Software/ Firmware
Microsoft Unified Communications, Rel. v3.0.6362 (continued)	OCS-FE1	HP Proliant BL460c	Microsoft Windows 2003 SP2
			Microsoft OCS 2007
			Office 2003 Web Components
			Microsoft Visual J# 2.0
			Microsoft Visual C++
			Microsoft IIS 6.0
			Tumbleweed 4.10.0.344
			ActivClient CAC 6.1 x86
	OCS-FE2	HP Proliant BL460c	Microsoft Windows 2003 SP2
			Microsoft OCS 2007
			Microsoft .NET Framework 2.0 SP1
			Microsoft Office 2003 Web Components
			Microsoft Visual C++
			Microsoft Visual J# 2.0
			Microsoft IIS 6.0
			Tumbleweed 4.10.0.344
			ActivClient CAC 6.1 x86
	OCS-BE	HP Proliant BL460c	Microsoft Windows 2003 SP2
			Microsoft SQL Server 2005 SP3
			Microsoft .NET Framework 2.0 SP1
			Microsoft Office 2003 Web Components
			Microsoft Visual Studio 2005
			Tumbleweed 4.10.0.344
			ActivClient CAC 6.1 x86
	AudioCodes Mediant 1000	T1 Trunks	F580A.042.003
		FXS (x2)	
		CPU	
	OCS-MED1	HP Proliant BL460c	Microsoft Windows 2003 SP2
			Microsoft OCS 2007 v3.5.6907.0
			Microsoft .NET Framework 2.0 SP2
			Tumbleweed 4.10.0.344
			ActivClient CAC 6.1 x86

SUT Telephones

Telephone type	Model
2-Wire Analog	Panasonic KX-T105W (See note.)
2-Wire Analog	Panasonic KX-T105W (See note.)

NOTE: The SUT is certified with any 2-Wire analog instrument that meets FCC Part 15 and Part 68 requirements.

LEGEND:

5ESS	Class 5 Electronic Switching System	ISA	Internet Security and Acceleration
AuthSer	Authentication Server	MED	Mediation
BE	Back End	NA	Not Applicable
CAC	Common Access Card	OCS	Office Communications Server
CPU	Central Processing Unit	RADIUS	Remote Authentication Dial In User Service
CS	Communication Server	Rel.	Release
FE	Front End	SP	Service Pack
FXS	Foreign Exchanges Station	SUT	System Under Test
HP	Hewlett-Packard	SQL	Structured Query Language
IAS	Internet Authentication Services	T1	Digital Transmission Link Level 1
IIS	Internet Information Services	v	version
IOS	Internetwork Operating System	XP	Experience

10. TESTING LIMITATIONS. None.

11. TEST RESULTS

a. Discussion

(1) DSN Trunk Interfaces. The SUT met all critical CRs and FRs for T1 ISDN PRI National ISDN (NI) 1/2 (American National Standards Institute [ANSI] T1.607) and E1 ISDN PRI (International Telecommunication Union – Telecommunication Standardization Sector [ITU-T] Q.931).

(2) DSN Line Interfaces. The SUT supports 2-Wire Loop Start Analog and VoIP softphones. The SUT does not support VoIP hard phones. The SUT met all critical interoperability certification requirements for 2-Wire Loop Start Analog (GR-506-CORE) and VoIP DSN line interfaces with the following minor exceptions:

(a) The SUT 2-Wire analog interface is provided by their Audio Codes Mediant 1000 gateway. Due to interoperability interaction problems with line features supported by this gateway, the line features (i.e. call waiting, call hold, call transfer etc.) are disabled with a software patch and not authorized for use. Line features for a PBX 2 are not required; therefore the operational impact is minor.

(c) The SUT VoIP interface met all critical CRs and FRs with the following exception: During testing, there were two occasions when all softphones lost registration with the call controller and multiple occasions where individual softphones lost registration. When these failures occurred, the softphones logged out from the controller and lost active communications. Call processing was lost for a period of time, up to 10 minutes, with no known explanation, but did reregister and become active. DISA adjudicated this discrepancy as having a minor operational impact.

(3) Features and Capabilities

(a) Common Features. The SUT met all critical interoperability certification requirements for Common Features with the following minor exception: Due to interoperability interaction problems with line features supported by the Audio Codes Mediant 1000 gateway, the 2-Wire analog line features (i.e. call waiting, call hold, call transfer etc.) are disabled with a software patch and not authorized for use. The SUT supports line features on their softphones to include: transfer, call hold, 3 way conferencing, call waiting, and call forwarding. The SUT also supports other features not tested. There is no risk associated with not testing these other features supported by the SUT.

(c) Public Safety. The SUT met the only required Public Safety requirement for a PBX 2: basic 911.

(d) Call Processing. The SUT met all critical CRs and FRs with following minor exception: During testing, there were two occasions when all softphones lost registration with the call controller and multiple occasions where individual softphones lost registration. When these failures occurred, the softphones logged out from the controller and lost active communications. Call processing was lost for a period of time, up to 10 minutes, with no known explanation, but did reregister and become active. DISA adjudicated this discrepancy as having a minor operational impact.

(e) ISDN Services. ISDN Services are conditional for a PBX 2; however, the SUT offers an ISDN PRI interface and met the PRI Access, Call Control and Signaling requirements for this interface.

(f) Synchronization. The SUT meets the minimum requirement of line timing mode with their Audio Codes Mediant 1000 gateway which supports an internal clock of Stratum 4 or better.

(g) Security. Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (f).

(4) VoIP. The SUT is certified with any Assured Services Local Area Network (ASLAN) or non-ASLAN on the UC APL.

(a) VoIP System. The UCR, paragraph 5.2.12.8.2, outlines the requirements for the VoIP system. The VoIP system requirements encompass end-to-end VoIP requirements. The following paragraphs detail the results of the SUT VoIP solution.

1. Voice Quality. In accordance with the UCR 2008, paragraph 5.2.12.8.2.1, PBX 2 VoIP system calls shall have an average Mean Opinion Score (MOS) of at least 4.0 as measured in accordance with the E-Model. Additionally, VoIP end instruments shall not lose two or more consecutive packets (excluding signaling packets) in a five-minute period. The SUT did not meet the minimum MOS requirements as stipulated in the UCR 2008. The MOS measured by the Sage 935 was an average MOS of 3.93 with a low MOS of 3.19. The E-Model MOS scores were lower, and had an average of 1.20 with a minimum of 1.12. Subsequently, JITC requested clarification from the DISA Program Manager on softphone requirements due to a disparity in UCR 2008 softphone requirements. The DISA Program Manager responded with the direction to exclusively use the UCR 2008, Change 1, Section 5.3.2.6.1.7 requirements for softphones with the intent to update the next issue of UCR. Therefore, based on UCR 2008, Change 1, Section 5.3.2.6.1.7, softphones are exempt from performance requirements which includes MOS, jitter, latency, and packet loss. The low MOS score was adjudicated by DISA as having a minor operational impact. Since the softphones do not provide tagging, they need to be connected directly to the Layer 3 switch, which will provide IEEE 802.1 p/Q VLAN tags, before connecting to a LAN as depicted in Figure 2-2.

2. Codec. In accordance with the UCR, paragraph 5.2.12.8.2.2, the ITU-T G.711 Pulse Code Modulation (PCM) CODEC with a 20 ms packet fill is required and was met by the SUT VoIP solution.

3. Security. Security requirements in accordance with the UCR, paragraph 5.2.12.8.2.4, are verified using the Information Assurance Test Plan. Results of the security testing are reported in a separate test report generated by the DISA Information Assurance test personnel, Reference (f).

4. Network Management (NM). In accordance with the UCR, paragraph 5.2.12.8.2.5, this is a conditional requirement for a PBX 2 was therefore not tested.

5. Synchronization. In accordance with the UCR, paragraph 5.2.12.8.2.6, the VoIP system shall meet all synchronization requirements identified in UCR, paragraph 5.2.10. The SUT derived synchronization with line timing mode via traditional T1 TDM-based interfaces and supports an internal stratum 4 clock.

6. Latency. The UCR, paragraph 5.2.12.8.2.7, states that one-way system latency for the VoIP system must be 60 ms or less as averaged over any five-minute period. The latency requirement is measured from IP and analog handsets to the egress trunk. The SUT used Softphones running under a Windows XP and Vista operating systems. Per UCR 2008, Change 1, section 5.3.2.6.1.7, softphones are exempt from performance requirements which include latency. Latency from the analog end instruments on the Audiocodes gateway measured 58 ms between the analog handsets T1 or E1 egress which meets this requirement.

7. Internet Protocol version 6 (IPv6). In accordance with UCR, section 5.3.5, all VoIP systems submitted for testing must be IPv6 capable. Dual Stack solutions are preferred and tunneling solutions are unacceptable. IPv6 Capable-products, in accordance with UCR, section 4.3.1.3, can create or receive, process, and send or forward (as appropriate) IPv6 packets in mixed Internet Protocol version 4 (IPv4)/IPv6 environments. IPv6 capable products shall be able to interoperate with other IPv6 capable products on networks supporting only IPv4, only IPv6, or both IPv4 and IPv6, and shall also:

a. Conform to the requirements of the Department of Defense (DoD) IPv6 Standard Profiles for IPv6 Capable Products document contained in the DoD Information Technology Standards Registry (DISR).

b. Possess a migration path and/or written commitment to upgrade from the developer (company Vice President or equivalent) as the IPv6 standard evolves.

c. Ensure product developer IPv6 technical support is available.

d. Conform to National Security Agency (NSA) and/or Unified Cross Domain Management Office requirements for Information Assurance products.

The UCR 2008, Change 1 updated the rules of engagement for VoIP systems. In accordance with the UCR 2008, Change 1, Section 5.3.5.3 “Interim UC IPv6 Rules of Engagement”, a VoIP system must, at minimum, support dual stack IPv6 with its call control agent and IP end instruments; however, softphones are exempt from IPv6. Since the only IP end instrument provided by the SUT is a softphone, IPv6 is not applicable. The SUT does not support IPv6.

8. In accordance with the UCR 2008, Section 5.2.12.8.2.9 and UCR 2008, Change 1, IPv6 interim rules of engagement a VoIP system call control agent and VoIP end instruments (excludes softphones) shall support IPv6 dual stack. The SUT VoIP system requirements in the paragraphs below shall be met. In order to meet the Quality of Service requirements the SUT includes two Cisco Catalyst 3560G “edge” and “core” switches. The SUT is certified with these switches or any other layer 3 access switches listed on the UC APL.

a. IP components shall be capable of implementing Service Class tagging using the 6-bit traffic class in the IPv6 header (excludes softphones, and analog phones on media gateways) and DSCPs field in the IPv4 header. The SUT is capable of implementing DSCP tagging in the IPv4 header only, which meets this requirement.

b. IP components shall be capable of assigning DSCP (0-63) to any distinct service class for traffic that traverses the device in accordance with UCR, Table 5.3.1-3. In accordance with the UCR, the DSCP field of the IP traffic associated with the distinct service classes of the session control components can be assigned a unique value by the SUT, which meets this requirement for IPv4 only.

c. For VoIP, video, and data end products, any end system that supports convergence (i.e., more than one media) the end-system must preassign the virtual LAN (VLAN) using Institute of Electrical and Electronics Engineers (IEEE) 802.1Q tags prior to the frames entering the ASLAN in accordance with UCR, section 5.3.1.7.4. For end-systems that support just one media (i.e., voice or video or data), the LAN can assign the VLAN based on port-based VLAN assignment. The SUT VoIP session control components used port based VLAN assignment, which meets this requirement.

9. Softphone Requirements. The SUT utilized two Softphones in the test. One softphone ran under Windows XP, the other under Windows Vista. All softphones used the soundcard in the computers to accept microphone input and generate speaker output. Since the softphones do not provide tagging, they need to be connected directly to the Layer 3 switch, which will provide IEEE 802.1 p/Q VLAN tags, before connecting to a LAN as depicted in Figure 2-2. The softphone had dual Ethernet connections. One Ethernet connection was used for data only and the other interface was used for voice only. Voice and data traffic has different tagging requirements. The operating system is point and click. Per DISA interim guidance provided to JITC on

8 April 2010, the UCR 2008, Change 1, section 5.3.2.6.1.7, requirements are applicable to all softphones including softphones on legacy switches (i.e. PBX2, PBX1, SMEO etc.). In some instances as noted below the requirements are specific to LSCs and not applicable to the SUT. In accordance with UCR 2008, Change 1, Section 5.3.2.6.1.7, the softphone shall be conceptually identical to a traditional IP "hard" telephone and is required to provide voice features and functionality provided by a traditional IP hard telephone, unless explicitly stated here within this paragraph. The softphone application in conjunction with a general-purpose computer, including its mouse (point and click) interaction, shall support, as a minimum, the following UCR 2008 change 1 requirements:

a. Section 5.3.2.2.2.1, Voice Features and Capabilities. This section identifies assured services interact with line features and is not applicable to a PBX2. The SUT is not required to support any line features, however the only two features tested with the softphone were hold and transfer.

b. Section 5.3.2.5.2.1, System Availability. A Softphone is exempt from availability requirements.

c. Section 5.3.2.6.1, Voice Instrument. The SUT met all the feature requirement in this section minus the assured services requirements which are not required for a PBX2.

d. Section 5.3.2.6.1.1, Tones and Announcements. Tones and announcements, as required in UCR 2008, Sections 5.2.4.5.2, DSN Information Signals, and Section 5.2.2.1.3, Announcements, shall be supported, except for the loss of C2 announcement. In accordance with sections 5.2.4.5.2 and 5.2.2.1.3 PBX2s are not required to meet these requirements. These requirements are therefore conditional for the SUT.

e. Section 5.3.2.6.1.2, Audio Codecs. This requirement states that an IP end instrument support the following audio codecs: G.711, G.723.1, G.722.1, G.729 and G.729a, however, only the G.711 audio codec is applicable to the SUT which was met.

f. Section 5.3.2.6.1.3, Handset Loudness and Frequency Response Requirement VoIP PEI or AEI Telephone Audio Performance Requirements. The SUT recorded MOS scores lower than 4.0, however in accordance with UCR 2008 change 1 section 5.3.2.6.3 a softphone is no required to meet the end to end performance requirements which includes MOS, the SUT is exempt from this requirement.

g. Section 5.3.2.6.1.4, Voice over IP Sampling Standard. The SUT meets this requirement with the 20ms Codec sampling rate.

h. Section 5.3.2.6.3, End Instrument to ASLAN Interface Section 5.3.3, Network Infrastructure End-to-End Performance Requirements. The softphone application shall be exempt from the performance (i.e., packet loss, jitter, latency) requirements specified in Section 5.3.3, Network Infrastructure End-to-End Performance Requirements, e.g., the PEI/AEI 50-ms codec latency and the 20-ms de-jitter buffer latency. Softphones are exempt from this requirement.

j. Section 5.3.3.3.2, VVoIP Differentiated Services Code Point (DSCP). The SUT with the addition of a layer 3 switch meets this requirement of tagging DSCP in accordance with this section. The SUT is certified with the the Cisco Catalyst 3560G Layer 3 switch or any other Layer 3 switch listed on the UC APL.

k. Section 5.4, Information Assurance Requirements: Softphone security and all IA requirements are provided in UCR 2008, Section 5.4, Information Assurance Requirements. It should be noted that softphones are required to support the VLAN IA requirements. Security is tested by DISA-led Information Assurance test teams and published in a separate report, Reference (g).

(b) Scalability. The SUT is scalable to support over 10,000 soft phone users. The SUT was tested with a total of two softphones, so maximum scalability was not evaluated. The system can have multiple TDM gateways managed by the call control agent. However, the SUT was evaluated with a single Audiocodes TDM gateway, so potential gateway routing and interaction were not evaluated. The system is capable of supporting hard IP phones; however, previous testing resulted in excessive latency measurements between the hard IP phones and the TDM egress. Due to this limitation, no hard IP phones are certified for use with this system.

(5) Network Gateways. The SUT met all critical interoperability certification requirements for the Public Switched Telephone Network (PSTN) Network Gateways with the following interfaces: T1 ISDN PRI NI 1/2 (ANSI T1.607) and E1 ISDN PRI (ITU-T Q.931).

b. System Interoperability Results. The SUT is certified for joint use in the DSN as a PBX 2 in accordance with the requirements set forth in the UCR. The SUT was tested and is certified for use with analog and VoIP softphones (computers emulating telephones) only. The SUT was not tested and is not certified with VoIP hard phones (traditional desktop VoIP phones). The interoperability test summary is shown in Table 2-3.

Table 2-3. SUT Interoperability Test Summary

DSN Trunk Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
T1 CAS (DTMF, MFR1, DP)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
E1 CAS (DTMF, MFR1, DP)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
T1 ISDN PRI NI 1/2 (ANSI T1.607)	Yes	Certified	Met all critical CRs and FRs.
E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRs.
DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
2-Wire Analog (GR-506-CORE)	Yes	Certified	Met all critical CRs and FRs with the following exceptions: The SUT 2-Wire analog interface is provided by their Audio Codes Mediant 1000 gateway. Due to interoperability interaction problems with line features supported by this gateway, the line features (i.e. call waiting, call hold, call transfer etc.) are disabled with a software patch and not authorized for use. Line features for a PBX 2 are not required; therefore the operational impact is minor.
ISDN BRI NI 1/2 (ANSI T1.619a)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
2-Wire Proprietary Digital	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
VoIP (Softphone only) (Ethernet IEEE 802.3u)	No	Certified	The SUT only supports softphones, it does not support VoIP hard phones. The SUT met all critical CRs and FRs with the following exception: During testing, there were two occasions when all softphones lost registration with the call controller and multiple occasions where individual softphones lost registration. When these failures occurred, the softphones logged out from the controller and lost active communications. Call processing was lost for a period of time, up to 10 minutes, with no known explanation, but did reregister and become active. DISA adjudicated this discrepancy as having a minor operational impact.
Common Features	Yes	Certified	Met all critical CRs and FRs with the following minor exception: Due to interoperability interaction problems with line features supported by the Audio Codes Mediant 1000 gateway, the 2-Wire analog line features (i.e. call waiting, call hold, call transfer etc.) are disabled with a software patch and not authorized for use. The SUT supports line features on their softphones to include: transfer, call hold, 3 way conferencing, call waiting, and call forwarding. The SUT also supports other features not tested. There is no risk associated with not testing these other features supported by the SUT.
Public Safety	Yes	Certified	The SUT met the only required Public Safety requirement for a PBX 2: basic 911.
Call Processing	Yes	Certified	Met all critical CRs and FRs with following minor exception: During testing, there were two occasions when all softphones lost registration with the call controller and multiple occasions where individual softphones lost registration. When these failures occurred, the softphones logged out from the controller and lost active communications. Call processing was lost for a period of time, up to 10 minutes, with no known explanation, but did reregister and become active. DISA adjudicated this discrepancy as having a minor operational impact.
ISDN Services	No	Certified	ISDN Services are conditional for a PBX 2; however, the SUT offers an ISDN PRI interface and met the PRI Access, Call Control and Signaling requirements for this interface.
Synchronization	Yes	Certified	Met all critical CRs and FRs. The SUT meets the minimum requirement of line timing mode with their Audio Codes Mediant 1000 gateway which supports an internal clock of Stratum 4 or better.
Security	Yes	Certified	See note.

Table 2-3. SUT Interoperability Test Summary (continued)

DSN Line Interfaces			
Interface & Signaling	**Critical**	**Status**	**Remarks**
VoIP System	No	Certified	The SUT only supports softphones; it does not support VoIP hard phones. The SUT is certified for VoIP specifically with any certified ASLAN or non-ASLAN posted on the UC APL. In order to meet the Quality of Service requirements the SUT includes two Cisco Catalyst 3560G "edge" and "core" switches. The SUT is certified with these switches or any other layer 3 access switches listed on the UC APL.
Softphone	No	Certified	The SUT only supports softphones, it does not support VoIP hard phones. The SUT met all critical CRs and FRs with the following exception: During testing, there were two occasions when all softphones lost registration with the call controller and multiple occasions where individual softphones lost registration. When these failures occurred, the softphones logged out from the controller and lost active communications. Call processing was lost for a period of time, up to 10 minutes, with no known explanation, but did reregister and become active. DISA adjudicated this discrepancy as having a minor operational impact. Since the softphones do not provide tagging, they need to be connected directly to the Layer 3 switch, which will provide IEEE 802.1 p/Q VLAN tags, before connecting to a LAN as depicted in Figure 2-2.

Network Gateways				
Gateway	**Interface & Signaling**	**Critical**	**Status**	**Remarks**
PSTN	T1 CAS (DTMF, DP)	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	E1 CAS (DTMF, DP)	No (Europe only)	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.
	T1 ISDN PRI NI 1/2 (ANSI T1.607)	No	Certified	Met all critical CRs and FRS.
	E1 ISDN PRI (ITU-T Q.931)	No (Europe only)	Certified	Met all critical CRs and FRS.
	Ground Start Line	No	Not Tested	This interface is not supported by the SUT and is not required for a PBX 2.

NOTE: Security is tested by DISA-led Information Assurance test teams and published in separate reports, References (f) and (g).

LEGEND:

802.3u	Standard for carrier sense multiple access with collision detection at 100 Mbps
ANSI	American National Standards Institute
ASLAN	Assured Services Local Area Network
BRI	Basic Rate Interface
C2	Command and Control
CAS	Channel Associated Signaling
CRs	Capability Requirements
DISA	Defense Information Systems Agency
DP	Dial Pulse
DSN	Defense Switched Network
DSS1	Digital Subscriber Signaling 1
DTMF	Dual Tone Multi-Frequency
E1	European Basic Multiplex Rate (2.048 Mbps)
FRs	Feature Requirements
GR	Generic Requirement
GR-506-CORE	LSSGR: Signaling for Analog Interfaces
IEEE	Institute of Electrical and Electronics Engineers
ISDN	Integrated Services Digital Network
ITU-T	International Telecommunication Union - Telecommunication Standardization Sector
LSSGR	Local Access and Transport Area (LATA) Switching Systems Generic Requirements
Mbps	Megabits per second
MFR1	Multi-Frequency Recommendation 1
MLPP	Multi-Level Precedence and Preemption
MOS	Mean Opinion Score
NI 1/2	National ISDN Standard 1 or 2
PBX 2	Private Branch Exchange 2
PRI	Primary Rate Interface
PSTN	Public Switched Telephone Network
Q.931	Signaling Standard for ISDN
SS7	Signaling System 7
SUT	System Under Test
T1	Digital Transmission Link Level 1 (1.544 Mbps)
T1.607	ISDN Layer 3 Signaling Specification for Circuit Switched Bearer Service for DSS1
T1.619a	SS7 and ISDN MLPP Signaling Standard for T1
VoIP	Voice over Internet Protocol

12. TEST AND ANALYSIS REPORT. No detailed test report was developed in accordance with the Program Manager's request. JITC distributes interoperability information via the JITC Electronic Report Distribution (ERD) system, which uses Unclassified-But-Sensitive Internet Protocol Router Network (NIPRNet) e-mail. More comprehensive interoperability status information is available via the JITC System Tracking Program (STP). The STP is accessible by .mil/gov users on the NIPRNet at https://stp.fhu.disa.mil. Test reports, lessons learned, and related testing documents and references are on the JITC Joint Interoperability Tool (JIT) at http://jit.fhu.disa.mil (NIPRNet), or http://199.208.204.125 (SIPRNet). Information related to DSN testing is on the Telecom Switched Services Interoperability (TSSI) website at http://jitc.fhu.disa.mil/tssi. Due to the sensitivity of the information, the Information Assurance Accreditation Package (IAAP) that contains the approved configuration and deployment guide must be requested directly through government civilian or uniformed military personnel from the Unified Capabilities Certification Office (UCCO), e-mail: ucco@disa.mil.